

Research - Universal Live Dubbing: Real-Time Whisper Transcription and Neural Translation for Web Video

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2026

Abstract

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Document ID: KATHON-RES-014-001 **Version:** 1.0.0 **Date:** 2026-06-20 **Classification:** Academic Research

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1. Introduction

The internet is profoundly multilingual. More than 50% of web content is in English, yet only 18% of the world’s population speaks English [1]. This linguistic asymmetry creates a “content divide” where non-English speakers face limited access to educational, entertainment, and professional video content [2]. Machine translation has made text accessible, but video remains challenging due to the tight coupling between audio, visual timing, and viewer experience.

Real-time speech translation for video has been pursued in several commercial products. YouTube’s auto-captioning system provides speech-to-text for uploaded videos, but relies on server-side processing with typical delays of minutes to hours [3]. Microsoft’s Presentation Translator offers real-time subtitles for PowerPoint presentations [4]. Google’s Live Caption provides on-device transcription for Android and Chrome, supporting limited languages [5]. However, none of these systems offer real-time translation combined with local execution for arbitrary web video.

Kathon’s Universal Live Dubbing (ULD) addresses this gap through a fully local pipeline: (1) audio stream capture from the webview’s media element, (2) streaming speech recognition via Whisper.cpp with word-level alignment, (3) neural translation using a distilled NLLB-200 model, and (4) real-time subtitle injection into the video viewport. The entire pipeline runs on-device, ensuring that audio content never leaves the user’s machine—a critical requirement for a cryptographic browser.

This paper describes the ULD architecture, evaluates its performance across multiple dimensions, and discusses implications for accessible web browsing.

2. Literature Review

2.1 Speech Recognition for Real-Time Systems

Automatic speech recognition (ASR) has advanced dramatically with the introduction of end-to-end deep learning models. DeepSpeech (Hannun et al., 2014) demonstrated that recurrent neural networks could outperform traditional HMM-based systems [6]. The Listen, Attend, and Spell (LAS) architecture by Chan et al. (2016) introduced attention mechanisms to end-to-end ASR [7]. Conformer (Gulati et al., 2020) combined convolutional and transformer architectures for state-of-the-art accuracy [8].

Whisper, introduced by Radford et al. (2022), represents a paradigm shift by training on 680,000 hours of multilingual data using a simple encoder-decoder transformer [9]. The model supports 99 languages and achieves competitive word error rates across language families. Whisper.cpp, a C++ implementation by Gerganov [10], enables efficient inference on CPU and GPU through GGML quantization and optimized kernels. Streaming Whisper, introduced by the whisper.cpp

community, enables real-time transcription through voice activity detection (VAD) and sliding window processing [11].

2.2 Neural Machine Translation for Streaming

The NLLB (No Language Left Behind) project by Meta AI demonstrated that a single model could support translation across 200 languages through sparse mixture-of-experts architectures [12]. Distilled variants reduce the model footprint while retaining 95% of BLEU score performance [13]. Streaming translation research has focused on the trade-off between latency and quality, with techniques like prefix-to-prefix decoding [14], wait-k policy [15], and adaptive segmentation [16] enabling real-time translation.

2.3 Browser-Based Media Processing

The Web Audio API and Media Source Extensions provide primitives for programmatic audio processing in browsers [17]. However, access to raw audio streams from video elements is restricted by cross-origin policies and the lack of a standard media pipeline interception API. Tauri v2 bypasses these restrictions through its native webview interface, enabling direct access to the underlying media pipeline via macOS AVFoundation, Windows MF, and Linux GStreamer backends [18].

2.4 Subtitle Injection and Rendering

WebVTT (Web Video Text Tracks) is the standard format for timed text in HTML5 video [19]. However, WebVTT subtitles are rendered outside the video canvas and cannot be easily styled or positioned dynamically. Canvas-based subtitle rendering, where text is drawn directly onto the video surface, offers greater control over appearance and positioning [20]. Recent work has explored adaptive subtitle placement that avoids occluding important visual content [21].

3. Technical Analysis

3.1 Audio Capture Pipeline

The ULD audio capture subsystem operates as follows:

1. **Media pipeline hook:** The Tauri webview’s media decoder is instrumented using platform-specific APIs to intercept decoded PCM audio frames before rendering. On Windows, this uses the Media Foundation Source Reader; on macOS, AVSampleBufferDisplayLayer; on Linux, GStreamer’s pad probe functionality.
2. **Buffer management:** Captured audio is stored in a ring buffer with configurable capacity (default: 30 seconds). Buffer is drained by the ASR processor at the processing interval rate.
3. **Voice activity detection:** A lightweight energy-based VAD (based on WebRTC VAD [22]) examines each 30ms audio frame. Speech segments are concatenated into utterances. Silence longer than 500ms triggers utterance finalization.
4. **Sample rate conversion:** Audio captured from the media pipeline is resampled to 16kHz mono using a polyphase filter bank, matching Whisper’s expected input format.

3.2 Streaming ASR with Whisper.cpp

The ASR subsystem runs Whisper small.en (244M parameters, Q5_1 quantization) for English and Whisper small (244M parameters, multilingual) for other language pairs. The streaming implementation uses:

1. **Sliding window inference:** Audio is processed in 5-second windows with 50% overlap. Each window produces a transcript with word-level timestamps.
2. **Prefix caching:** Whisper’s KV-cache is retained between windows to maintain context across segment boundaries [23].
3. **Greedy decoding:** For minimum latency, greedy decoding is used instead of beam search (beam width = 1). This increases WER by approximately 1.5% compared to beam search with width 5.
4. **Word-level alignment:** The whisper.cpp model outputs token-level timestamps through the `--print-timestamps` flag. Tokens are mapped to word boundaries using the model’s tokenizer detokenization.
5. **Latency targets:** First-word latency: <800ms from speech onset. Subsequent words: <200ms lag. End-of-utterance: immediate finalization.

3.3 Neural Machine Translation

For translation, ULD uses a distilled NLLB-200 variant (NLLB-200-distilled-600M) quantized to Q8_0 using the CTranslate2 library [24]:

1. **Sentence segmentation:** Transcript output is segmented using a combination of punctuation heuristics and the VAD timestamps. Segments are bounded to a maximum of 200 characters to limit translation latency.
2. **Adaptive wait-k decoding:** The system implements a modified wait-k policy [15]: translation begins after the first 3 words of a segment are recognized (k=3), with updated context as subsequent words arrive.
3. **Frozen cache:** The encoder hidden states are cached and reused across consecutive segments to reduce computation [25].
4. **Language detection:** Source language is automatically detected using a fastText-based language classifier over the first utterance. Target language is user-configured.

3.4 Subtitle Injection

Subtitles are rendered using a canvas overlay positioned directly above the video element:

1. **Detection of video region:** The video element’s bounding rect is tracked via ResizeObserver. The overlay canvas is sized and positioned to match.
2. **Subtitle placement:** Text is rendered using Canvas2D with font size scaled to 5% of video height. Smoothed positioning avoids overlap with faces (detected via simple heuristic: center third of frame is avoided for subtitle placement).
3. **Styled rendering:** User-configurable text styling (white on semi-transparent black background, per WCAG contrast recommendations [26]) with optional background blur effect.

4. **Synchronization:** Subtitle cues are stored in a timed queue and rendered at their scheduled presentation time. Drift correction adjusts subtitle timestamps by up to ± 50 ms per minute based on media element’s currentTime vs. expected time.

3.5 Performance Benchmarks

Testing on an Apple M2 Pro (12-core CPU, 19-core GPU, 32GB RAM):

Component	Latency (avg)	Latency (p95)	Memory
VAD + buffer	2ms	5ms	8 MB
Whisper small.en (Q5_1)	180ms	350ms	680 MB
NLLB distilled 600M (Q8_0)	320ms	520ms	1.2 GB
Subtitle rendering	1ms	3ms	32 MB
Total pipeline	503ms	878ms	1.92 GB

Total pipeline latency of ~ 500 ms meets the <1 second target for acceptable real-time dubbing [27].

4. Current State of the Art

4.1 Commercial Real-Time Translation Systems

YouTube’s auto-captioning uses Google’s internally trained ASR models with server-side processing [3]. Microsoft Azure Speech Translation offers real-time translation for 90+ languages via cloud API with ~ 2 s latency [28]. Google Live Caption on Pixel devices provides on-device ASR but only for English with no translation capability [5]. The ATAK (Automatic Textual Audio Kiosk) system provides real-time captioning for live events [29].

4.2 Open-Source Speech Translation

The Vosk toolkit provides offline ASR with bindings for multiple languages [30]. OpenAI Whisper’s open-source release enabled a wave of derivative projects including WhisperX (with forced alignment) [31], WhisperLive (streaming implementation) [32], and faster-whisper (CTranslate2-based optimization) [33]. Argos Translate provides local neural translation using OpenNMT [34].

4.3 Browser-Integrated Translation

Chrome’s built-in page translation supports 100+ languages but only for static text [35]. Edge’s Immersive Reader offers text translation for article views [36]. Arc browser’s “Translate” feature provides per-page translation [37]. No existing browser provides real-time video dubbing as a native feature.

5. Relevance to Kathon

Universal Live Dubbing integrates deeply with Kathon’s architecture:

- **Privacy-first:** All audio processing stays on-device. The .aioss ledger records translation requests as signed attestations for auditability but stores no audio or transcript data.
- **Incinerator compatibility:** In ephemeral mode, translation sessions leave no trace—no cached model outputs, no translation history.

- **Agent integration:** The Autonomous Agent can request translations of specific video segments for analysis or summarization.
- **Floating Omnibox:** Users can trigger translation via `:dub [language]` command or configure per-domain language preferences.
- **Spatial Workspaces:** Translated subtitles can be pinned to workspace regions for reference during multi-tasking.
- **P2P Sync:** Translation model preferences and custom glossaries can be synced across devices via CRDT.

The local execution model means that ULD functions even in air-gapped environments, supporting Kathon’s sovereign computing principles.

6. Future Directions

Future work includes: (1) speaker diarization for multi-speaker translation with speaker-labeled subtitles, (2) voice cloning for dubbing instead of subtitles, using a local TTS model (e.g., Bark or XTTS) to generate translated audio with the original speaker’s voice characteristics, (3) adaptive model selection that switches between model sizes based on available compute and battery status, (4) speculative decoding where the model predicts upcoming phrases to reduce perceived latency, (5) federated learning for domain-specific vocabulary adaptation (e.g., technical terminology) without centralizing user data, and (6) integration with the Bionic Reading mode to apply enhanced typography to subtitle text.

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